SIP PoE IP Phone

VIP-255PT

User's manual

Version 1.0

Copyright

Disclaimer

PLANET Technology does not warrant that the hardware will work properly in all environments and applications, and makes no warranty and representation, either implied or expressed, with respect to the quality, performance, merchantability, or fitness for a particular purpose.

PLANET has made every effort to ensure that this User’s Manual is accurate; PLANET disclaims liability for any inaccuracies or omissions that may have occurred.

Information in this User’s Manual is subject to change without notice and does not represent a commitment on the part of PLANET. PLANET assumes no responsibility for any inaccuracies that may be contained in this User’s Manual. PLANET makes no commitment to update or keep current the information in this User’s Manual, and reserves the right to make improvements to this User’s Manual and/or to the products described in this User’s Manual, at any time without notice.

If you find information in this manual that is incorrect, misleading, or incomplete, we would appreciate your comments and suggestions.

CE mark Warning

The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

Trademarks

The PLANET logo is a trademark of PLANET Technology. This documentation may refer to numerous hardware and software products by their trade names. In most, if not all cases, their respective companies claim these designations as trademarks or registered trademarks.

Revision

User's Manual for PLANET SIP PoE IP Phone:

Model: VIP-255PT

Rev: 1.0 (2008, June)

Part No. EM-VIP255PTV1.0

TABLE OF CONTENTS

Chapter 1 Introduction

Overview

Meeting the next-generation Internet telephony service demands, the PLANET VIP-255PT is an ideal solution for office / home use as well as installation for Internet Telephony Service Provider (ITSP). VIP-255PT is a SIP IP phone with 802.3af Power over Ethernet (PoE) LAN interface supported. The built-in Graphic LCD of the VIP-255PT is with blue backlight and support multi-language on both LCD and webpage. The VIP-255PT is the delivery platform for IP voice services that brings benefits from the VoIP technologies in your daily life. The ITSP can diagnose and configure the phone remotely and thus reduce the cost of service.

The VIP-255PT has additional rich features including support of Media push, SMS, online advertisement, news and voice mail and etc., which would increase ARPU for service providers. It also features self-contained, service-integrated, intelligent phone functions, and powerful voice processing. The VIP-255PT can effortlessly deliver toll voice quality equivalent to the regular SIP Protocol connections by utilizing cutting-edge Quality of Service, echo cancellation, comfort noise generation (CNG) and voice compensation technology. Meanwhile, the dual Ethernet interfaces on the IP Phone allow users to install in an existing network location without interfering with desktop PC network connections.

Product Features

- SIP 2.0 and 802.3af PoE
- Multi–Language Function
- Online Advertisement, and SMS Function
- GIPS voice engine embedded to generate stable and clear voice quality
- Voice Codec: G.711, G.729AB, G.726, iLBC or G.723.1
- Supports VAD, CNG, AEC, AGC and Volume adjustment.
- Large graphic LCD with blue backlight supports
- Call hold, call waiting, call forward, call transfer, 3-way conference, auto answer and Hotline settings
- Supports Caller ID/Name display and DND
- Supports phone book, speed dial, call list, dial plan, volume adjustment and rings selection
- Supports NAT transverse: STUN mode
- IP Assignment: Static IP/ DHCP/PPPoE
- Supports in-band DTMF and out-of band RFC2833 DTMF

- Supports Proxy mode and peer-to-peer SIP link mode
- Supports standard encryption and authentication (MD5 and MD5-sess)
- The phone can be configured via keypad, web browser or remote.
- Firmware can be upgraded through HTTP, FTP or TFTP.

Package Content

The contents of your product should contain the following items:

SIP PoE IP Phone Unit

Power adapter

Quick Installation Guide

User's manual CD

Physical Details

The following figure illustrates the front/rear panel of IP Phone.

Rear View

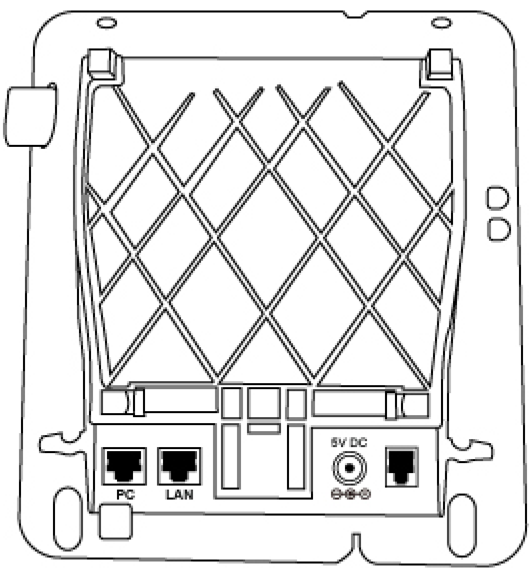


Figure 1. Rear Panel

PC	RJ-45 connector, to maintain the existing network structure, connected directly to the **PC** through **straight** CAT-5 cable
LAN	RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable. The **LAN** interface also can be connected with 802.3af PoE switch or converter for power supply.
DC 5V	5V DC Power input outlet
Handset	RJ-11 connector, connected directly to the Handset.

ⓘ**Note**

For VIP-255PT, either PoE or AC adapter can be deployed at one time

Front View and Keypad function

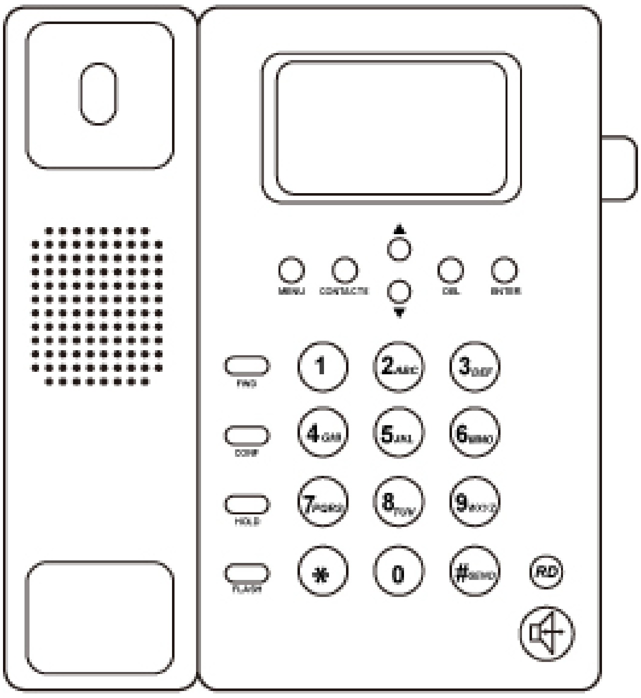


Figure 2. Ffront Panel

Keypad Description

LCD Display	Menu and all status shall be displayed for users.
MENU	To bring out the menu selection while IP Phone is in idle state.
▲ ▼	This is Up ▲ / Down ▼ key and volume setting when off-hook off. Show the calls history when on-hook.
ENTER	To be used as confirm configuration or enter sub-menu.
CONTACTS	Enter the phone book selection.
FLASH	To transfer an active call (incoming call answered or outgoing call accepted) to another devices.
CONF	Press this button can make conference function.
FWD	To carry out forward function.
DEL	Press to delete digits when at configuration mode or input phone numbers. Press to mute sounds when at talk mode.
RD	Press to dial the last dialed number when the IP Phone is off-hooked.
Handfree	To switch between the usage of the handset and the speaker devices.
Hold	To hold the conversation.

Icon on the LCD

When the phone is in different mode, the LCD display shows different icons.

Graphic Icon Description

	Network status icon: Flash in the case of Ethernet linking failure.
	Register status icon: fail to register to the server
	Missed calls
	All kinds of characters input mode icon, press **Contacts** key to select input method
	Digital input
	Small letter input
	Capital letter input
	Mute microphone
	Call held
	Voice mail
	SMS
	Always call forward
	Busy Call Forward
	No-answer Forward
	DND (Don't disturb)
	Auto Answer

Chapter 2

Preparations & Installation

Physical Installation

VIP-255PT: 802.3af PoE SIP IP Phone (2 x RJ-45, 1 x PoE for LAN interface)

Step 1: Connecting Handset

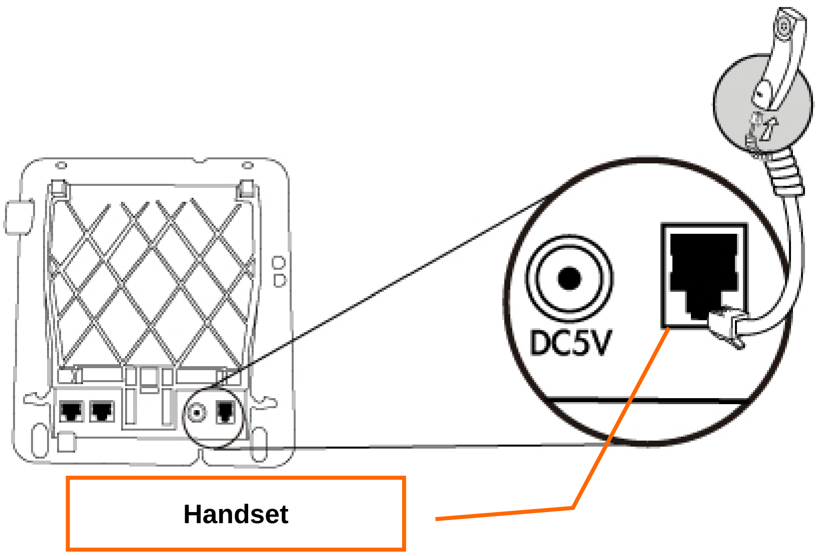
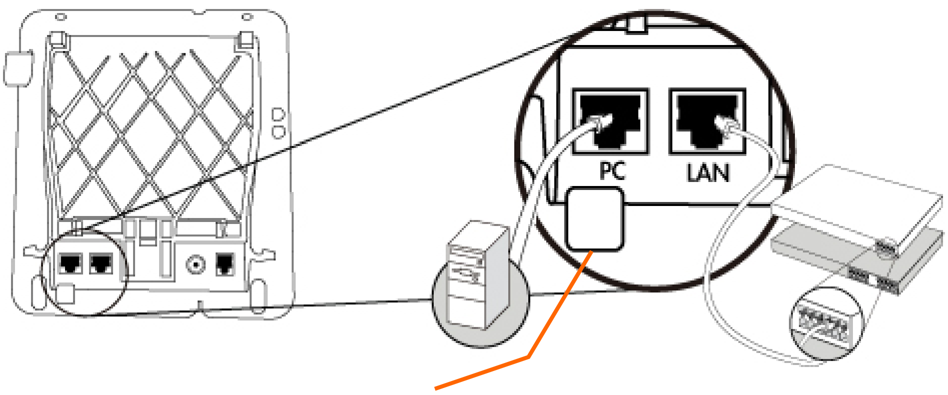


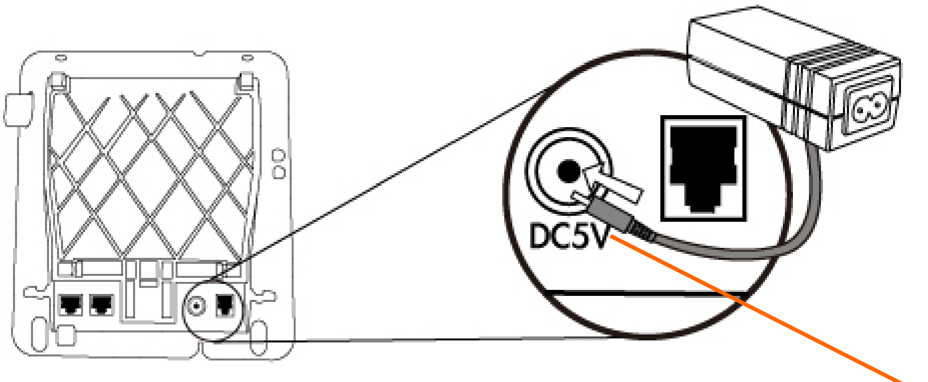
Figure 3 handset installation

Step 2: Connecting Power AC Power and Network



Plug the Ethernet cable into the back of the base station. Plug the other end of the Ethernet cable into your already prepared network connection.

Figure 4 LAN/PC port installations

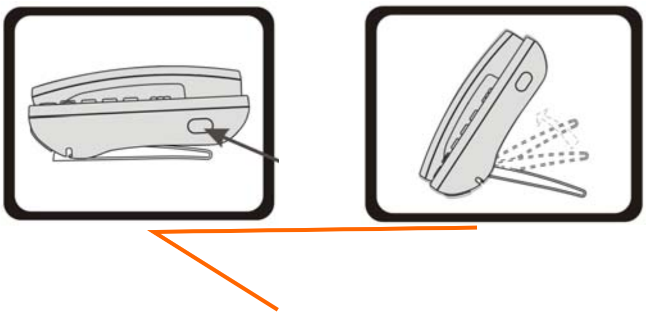


Power Adapter (5V DC)

NOTE: Use only the power adapter shipped with the unit to ensure correct functionality.

Figure 5 power adapter installations

Step 3: Adjust the stand angle.



Press and hold the button of right side to change the stand mount angle.

Figure 6 stand angle adjustment

Administration Interface

The IP Phone provides GUI (Web based, Graphical User Interface) for machine management and administration. Key pad administration also available for simple configuration.

Web configuration access

To start IP Phone web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.0.0 or higher with Java support

Default IP address of IP Phone is **192.168.0.1**. You may now open your web browser, and insert ***http://192.168.0.1*** in the address bar of your web browser to logon IP Phone web configuration page. IP Phone will prompt for logon username/password, please enter: ***root*** / **null** (no password) to continue machine administration.

Note

In order to connect machine for administration, please locate your PC in the same network segment (192.168.0.x) of IP Phone. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

Chapter 3

Network Service Configurations

Configuring and monitoring your IP Phone from web browser

The IP Phone integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

Manipulation of IP Phone via web browser

Log on IP Phone via web browser

After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon IP Phone web configuration page.

IP Phone will prompt for logon username/password: ***root*** / ***null (without password)***

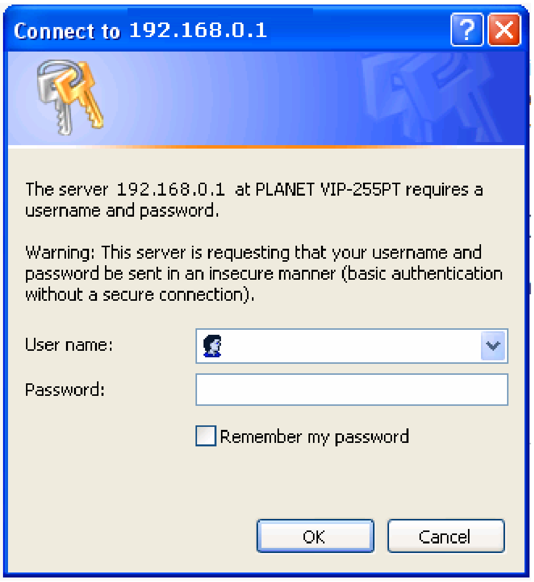


Figure 7. Login prompt page

When users login the web page, users can see the IP Phone system information like firmware version, company…etc in this main page.

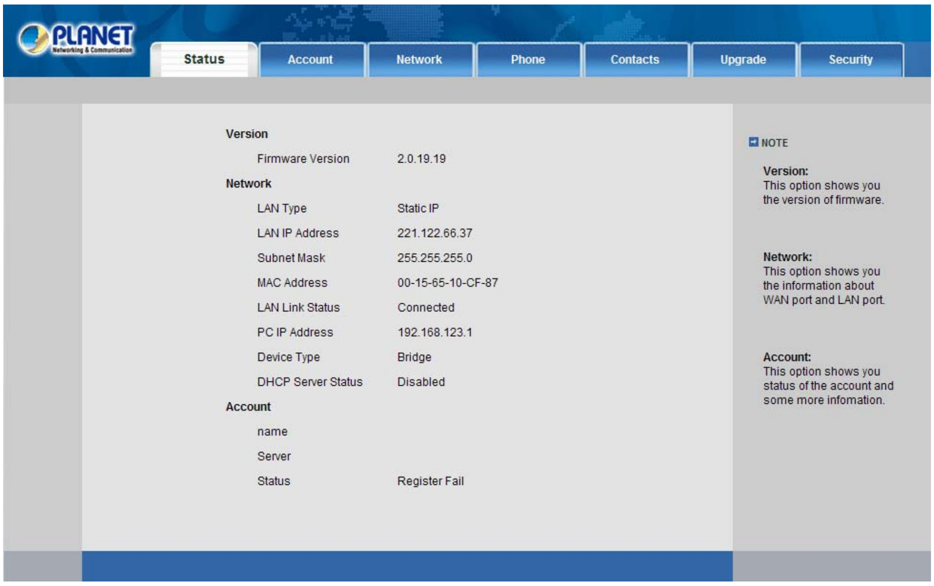


Figure 8 main page

Network configuration via web configuration interface

Execute your web browser, and insert the IP address (**default: 192.168.0.1**) of VIP-255PT in the address bar. After logging on machine with username/password (default**: root / no password**), browse to "**Network**" --> "**LAN Settings**" configuration menu:

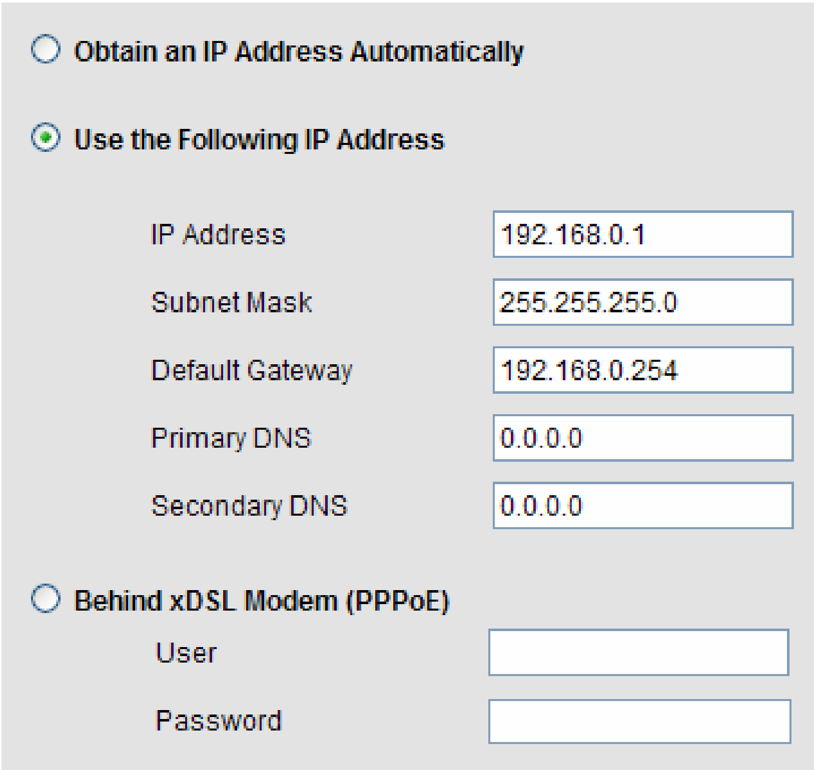


Figure 9. LAN port setting page

LAN Parameter Description

IP address	LAN IP address of IP Phone **Default**: 192.168.0.1
Subnet Mask	LAN mask of IP Phone **Default**: 255.255.255.0
Default Gateway	Gateway of IP Phone **Default**: 192.168.0.254

After confirming the modification you've done, please click on the **Confirm** button to apply settings and the machine will be reboot to make the settings effective.

Connection Type	Data required.
Obtain an IP Address Automatically	The ISP will assign IP Address, and related information.
Use the Following IP Address	In most circumstances, it is no need to configure the DHCP settings.
Behind xDSL Modem (PPPoE)	The ISP will assign PPPoE username / password for Internet access,

PC Port Parameter Description

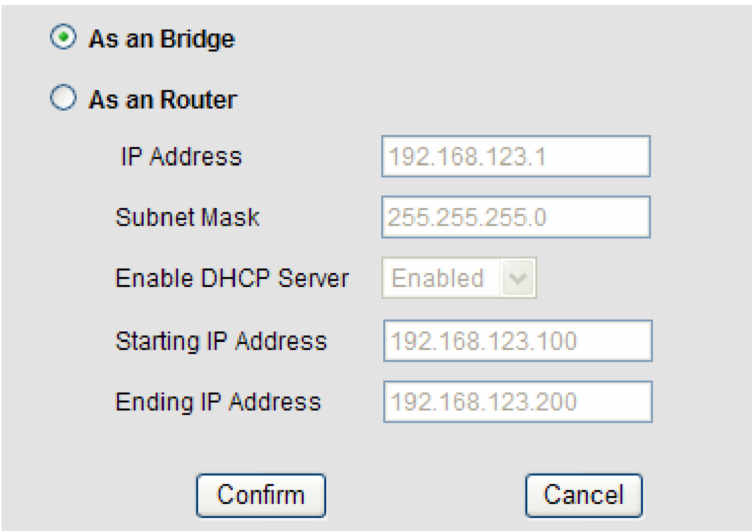


Figure 10. PC port setting page

Field Type	Description
Bridge	If you select the Bridge mode, then the two fast enternet port will be transparent.
Router	If you select the Router mode, the SIP phone will work as a router.

After confirming the modification you've done, Please click on the **Confirm** button to apply settings and the machine will be reboot to make the settings effective.

ⓘ Hint

```
Please consult your ISP personnel to obtain proper PPPoE/IP
address related information, and input carefully.
If Internet connection cannot be established, please check
the physical connection or contact the ISP service staff
for support information.
```

VLAN configuration

This page defines the VLAN setting in this page. This function needs to co-operate with network devices which have VLAN function.

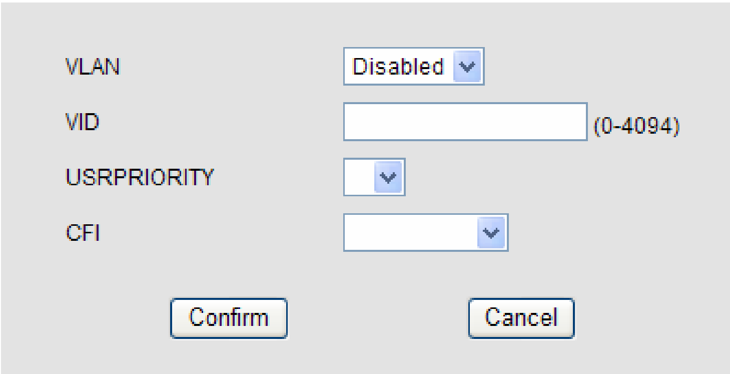


Figure 11. VLAN setting page

Field Type	Description
VID	Dispose VLAN ID is add a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 2~4094.
USRPRIORITY	To setup the user priority
CFI	To indicate the Canonical Format. • If Enable, it means the header label include RIF field, and the NCIF flag valus of RIF will to decide the MAC address is Canonical Format or Non-Canonical Format in frame information. • If Disable, it means the header label does not include RIF field, and the MAC address is Canonical Format in frame information.

Chapter 4

VoIP IP Phone Configurations

Baisc Function Configurations

Account Settings

In account information user need to input the account and the related informations in this page, please refer to your ISP provider.

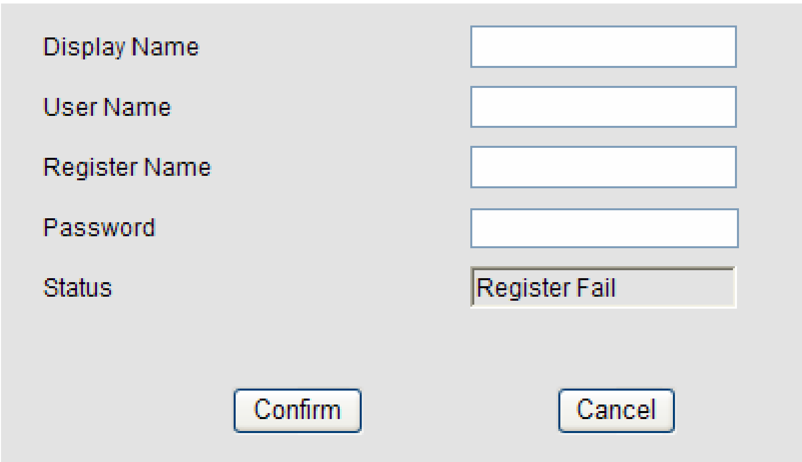


Figure 12. Account setting page

First of all, user need to input the following fields

Field	Description
Display Name	you can input the name you want to display
User Name	you need to input the User Name get from your ISP
Register Name	you need to input the Register Name get from your ISP
Register Password	you need to input the Register Password get from your ISP.

You can see the register status field. If the item shows “**Registered**”, indicated the IP Phone is registered to the ISP, user can make a phone call direcly.

User may get account information from your service provider. Press Confirm button to save the settings.

Registrer Server

In server information you need to input the register server and the related informations in this page, the same please refer to your ISP provider.

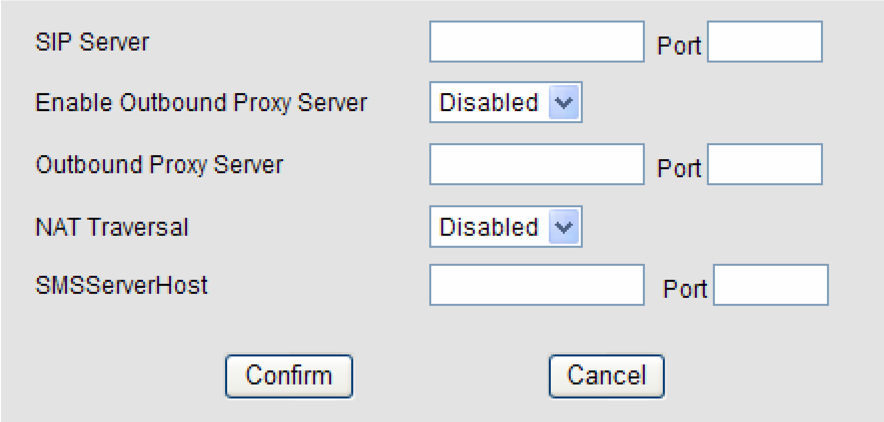


Figure 13. Register server setting page

Field	Description
SIP Server	you need to input the SIP Server get from your ISP
Enable Outbound Proxy Server	If your ISP does not provide the information, please disable this item.
Outbound Proxy Server	you need to input the Outbound Proxy get from your ISP.
NAT Traversal	The NAT Tranversal is Enable/Disable the STUN Server function in this parts that can help your VoIP phone working properly behind NAT. Change this settings please following your ISP provider.

Press Confirm button to save the settings.

Wait a moment for registering to the server, then return to Account page to check the register status. If it displays “**Registered**”, you can make calls now.

Voice Settings

This page defines the Codec priority, DTMF type, and VAD/CNG/Echo canceller function in this page. User need to follow the ISP suggestion to setup these items. When finished the setting, please click the Confirm button.

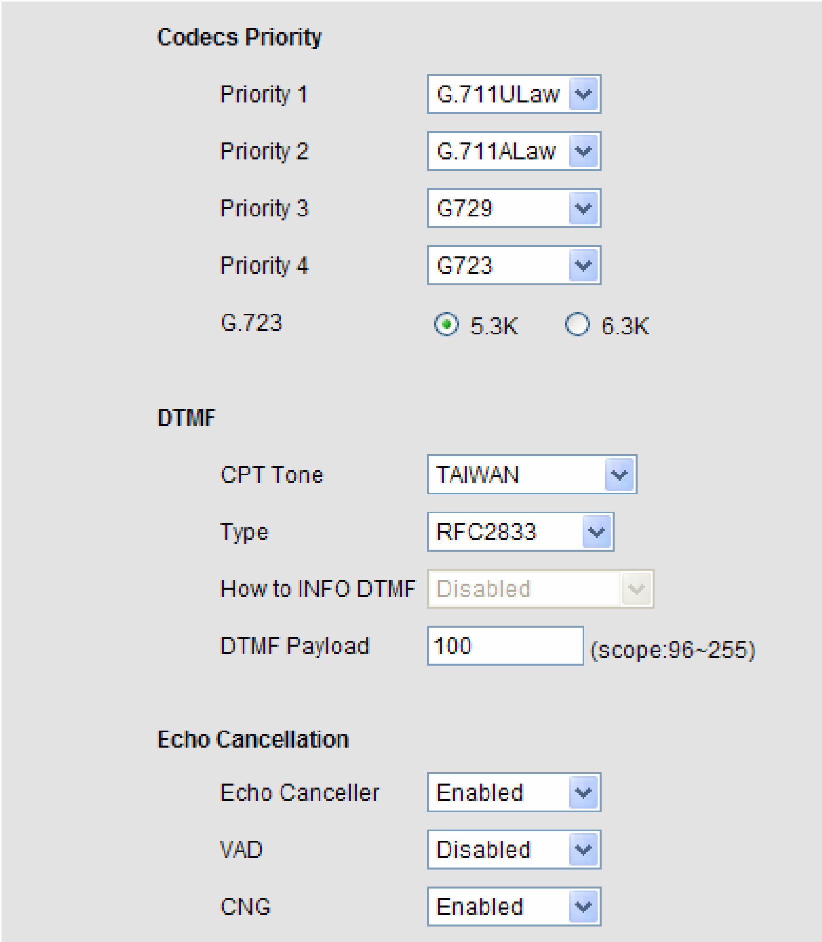


Figure 14. Voice setting page

Field	Description
Codec Priorities	There are 4 types of codec. User could select the priority of these codecs or set it to disabled, but at least you must select one type.
DTMF Payload Type	Sets the payload type for DTMF.
DTMF Payload	RTP payload for DTMF.

Advanced Settings

This page defines the advanced of account settings includes STUN server IP address, SIP/RTP port, SIP/Voice QoS setting and etc. Please click the confirm button to make effective when finished the setting.

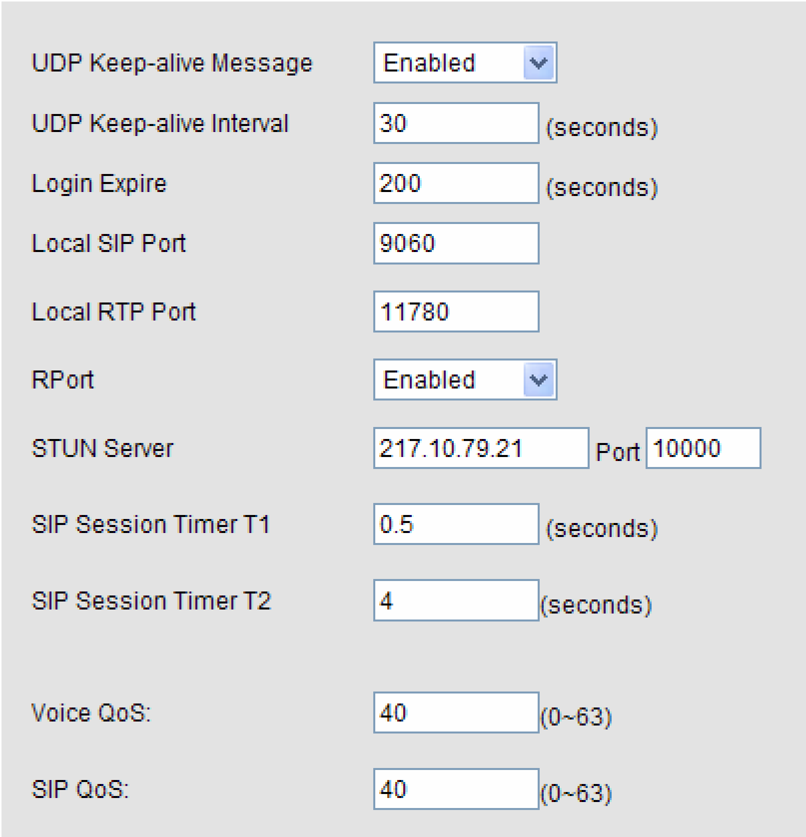


Figure 15. SIP advanced setting page

Field	Description
UDP Keep-alive Message	To deliver the packets on a regular time schedule to keep NAT port could open continued.
UDP Keep-alive Interval	To setup the schedule time for delivering the packets
Login Expire	This parameter allows user to specify the time frequency that unit refreshes its registration with the specified registrar.
Local SIP Port	To defines the SIP port number, please follow your ISP
Local RTP Port	To defines the RTP port number, please follow your ISP
RPort	The parameter allows SIP phone to tell the proxy to only send responses back to a particular address and port.
STUN Server	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
SIP Session Timer T1/T2	Allow you to trun on a timer to check if a SIP session is still active or should be terminated.
Voice/SIP QoS:	Enable the QoS feature configure the QoS ID values

Phone Preference Settings

This page defines the IP phone preference as language, ring type, advertisement, etc. Please click the confirm button to make effective when finished the setting.

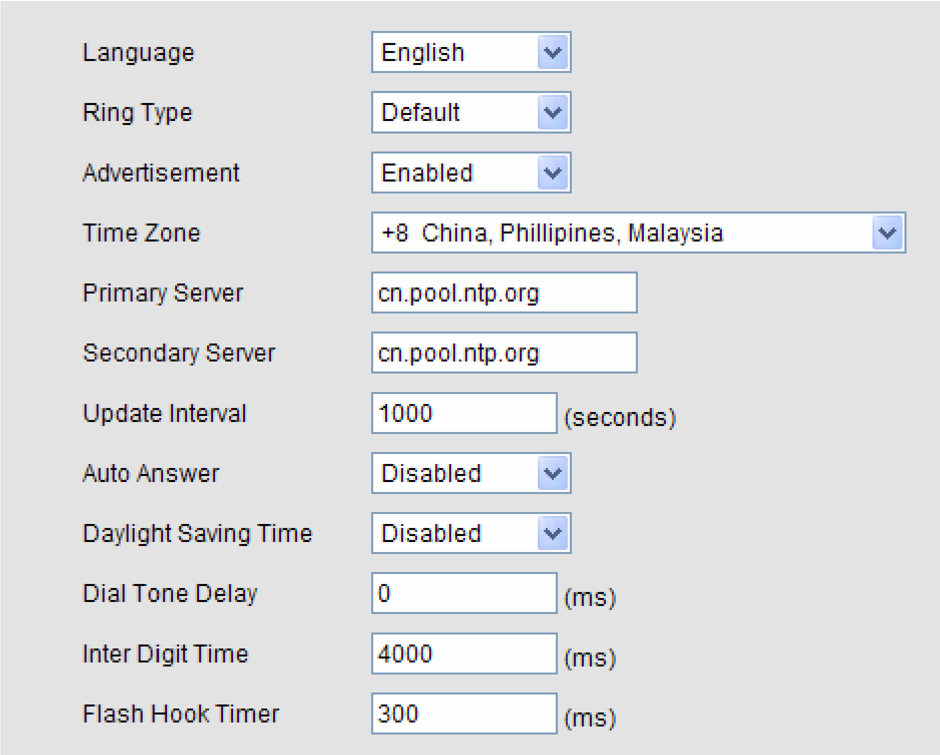


Figure 16. Phone preference setting page

Field	**Description**
Language	To defines the LCD display and webUI language of IP phone.
Ring Type	To defines the ring style.
Advertisement	Enable/Disable the LCD advertisement.
Time Zone	To defines base on your location to set the Time Zone.
Primary/Secondary Server	To the Primary and Secondary NTP server IP address.
Update Interval	To define how long need to synchronize again.
Auto Answer	Enable/Disable Auto-answer incoming calls arrive.
Daylight Saving Time	Enable/Disable the daylight saving time function.
Dial Tone Delay	To define the Dial Tone Delay time.
Inter Digit Time	To define the inter digit time.
Flash Hook Timer	To defince the time for user press the Hook to represent the Flash require.

Phone Function Settings

This page defines call forward, call waiting, voicemail number, hotline, programmable keys assign and etc. Please click the confirm button to make effective when finished the setting.

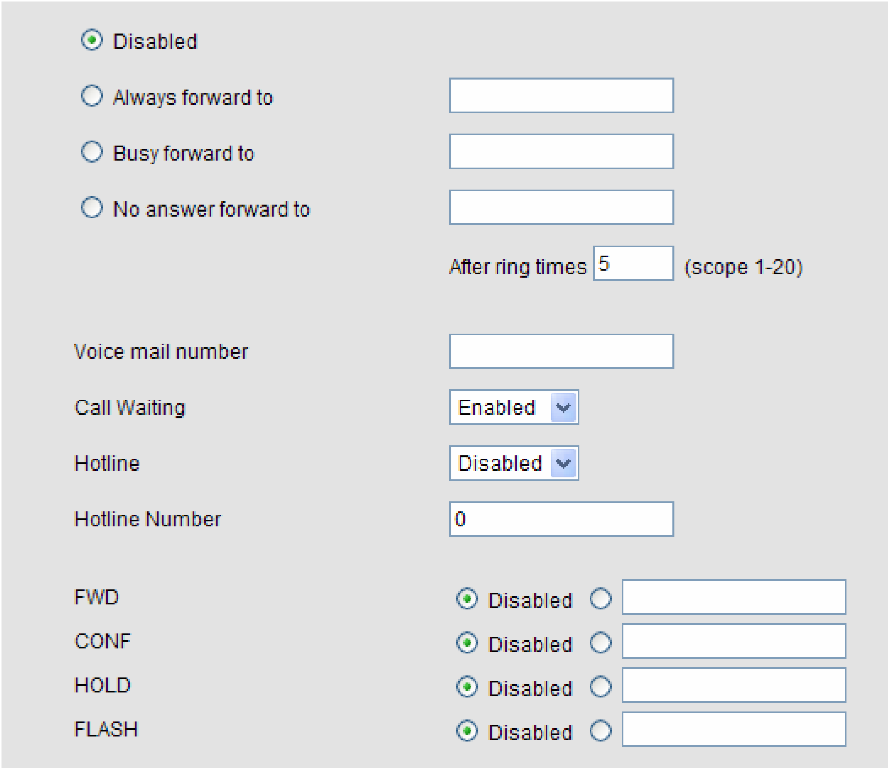


Figure 17. Phone function setting page

Field	**Description**
All forward	All incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.
Busy forward	If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.
No answer forward	If you can not answer the phone, the incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.
Voice Mail Number	Dial this number to access voice mail system, you cold get this number form you ISP
Call Waiting	If you disable this function, the secord incoming call will be

	declined when you are on the call.
Hotline	When you pick up the handset, your IP phone will dial the hotline number out automatically.
Programmable keys	These four keys can be configured as programmable keys. To use this function, you must first choose the radio box in front of the blank and input the assigned number in the blank. If you enable this function, the assigned number will be dialed out once you press this key, but the primal function will be lost at the same time.

Dial Plan Settings

Users could edit some dial plan by themselves. There are two kinds of rules, Replace Rule and Dial Now Rule.

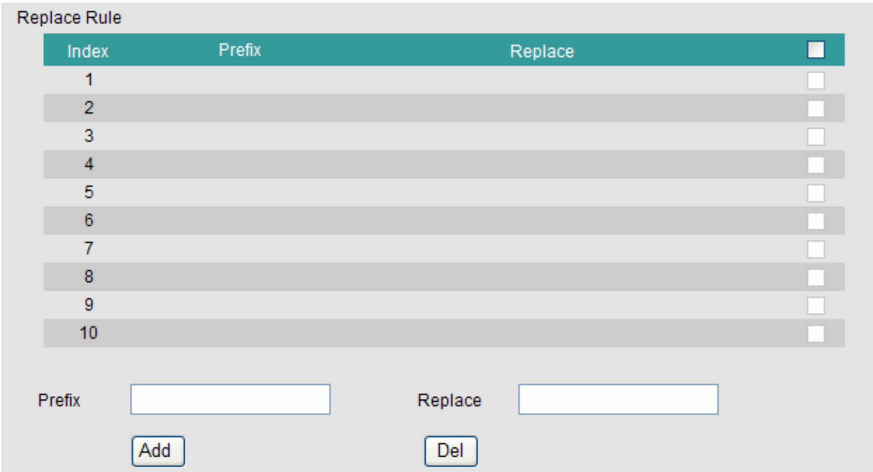


Figure 18. Replace Rule page

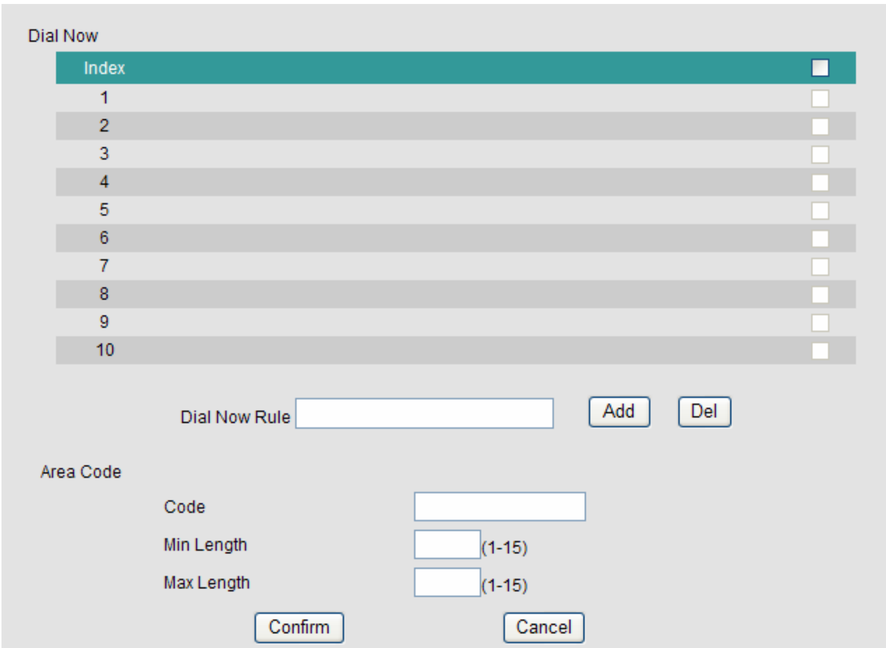


Figure 19. Dial now page

Replace Rule:

To define a rule to dial out with 'Replace' instead of 'Prefix'.

Dial Now:

The numbers could be dialed out immediately as long as it meet the rule user-defined.

For example:

- If you set prefix as 36 to replace 003136, when you press 36, it will be replaced by 003136.
- If you set prefix as 001 to replace 002, when you press 001, it will be replaced by 002.
- If you set dial now rule as xxxxxxxx, when you press 8 numbers such as 12345678, it will be dialed out immediately.
- If you set dial now rule as xxxx89, when you press 123489, 234589 etc., it will be dialed out immediately.

Edit SMS

Users could edit Short Messages send to other SIP phone through the SMS service.

Figure 20. SMS edit page

Contact Settings

Users could add/del/edit/search the contact list in this page; these numbers will also show on the contact list of LCD menu, that max up to 220 entries of contact.

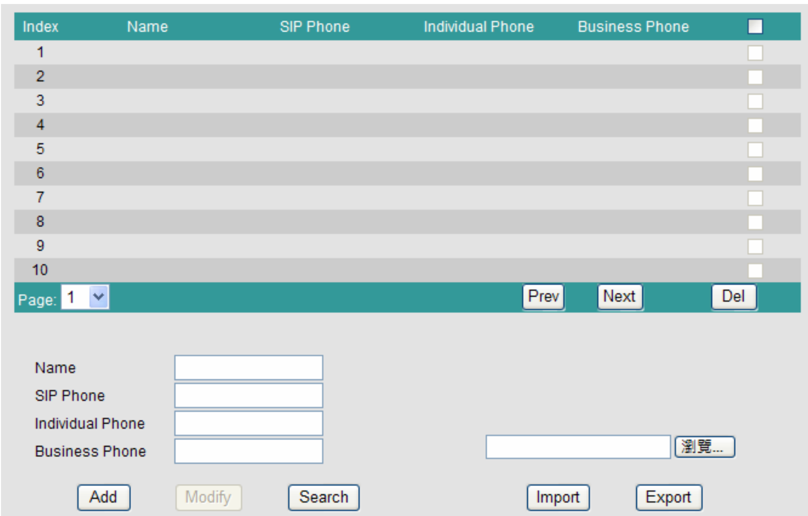


Figure 21. Contact list page

Field	Description
Page	The default is Page 1. It can select Page1 ~ Page 22 to look round Contact-List records.
Index	The record number from 1 ~ 10, it can set up 220 records in total.
Name	The name of contact records, it only can input numerals.
SIP / Individual / Business Phone	Fill in the outgoing number (Line Number) or IP address.

If you need to add a phone number into the contact list, you need to input the name, and the SIP phone number. When you finished a new contact list, just click the "**Add**" button.

If you want to delete a phone number, you can select the phone number you want to delete then click "**Del**" button.

If you need to edit a phone number, you can clieck the contact information in the table, then it will be displayed in the entry box, and then you could modify it and click the button "**Modify**" to submit.

If you want to delete all phone numbers, you can click the grid in the title and then click the "**Del**" button.

When you want to backup whole contact list, you could click the "**Export**" button and create a name which you want to store.

When you want to restore contact list, you could click the "**Browse**" button and select the contact (file in CVS format) you want to import, then click the "**Import**" button.

Speed Dial settings

In Speed Dial settings page you can add/delete Speed Dial number. You can input maximum 4 entries in this list. You can setup the Speed Dial number. If you want to use Speed Dial you just dial the speed dial number (from 0~99) and follow the "#" key.

If you need to add a phone number into the Speed Dial list, you need to input the Prefix and the Replace number. When you finished a new phone list, just click the "**Add**" button.

If you want to delete a phone number, you can select the phone number you want to delete then click "**Del**" button.

If you want to delete all phone numbers, you can click the grid in the title and then click the "**Del**" button.

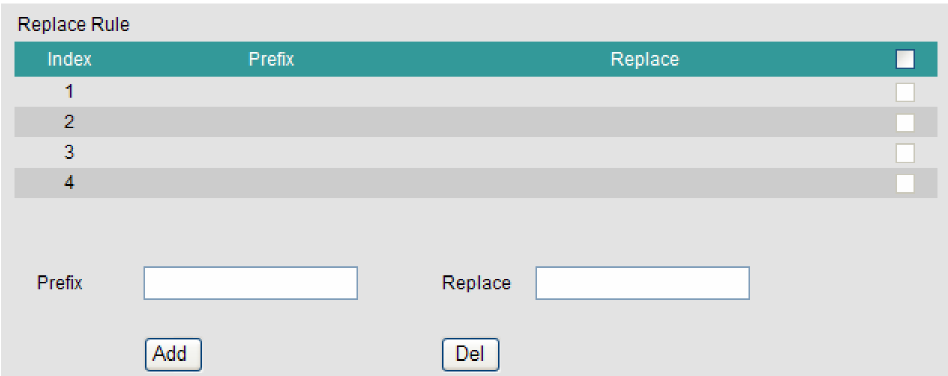


Figure 22. Speed-dial list page

Firmware Upgrade

This upgrade function page, you can run Rest settings to factory default, Reboot machine and Upgrade new firmware via HTTP in here.

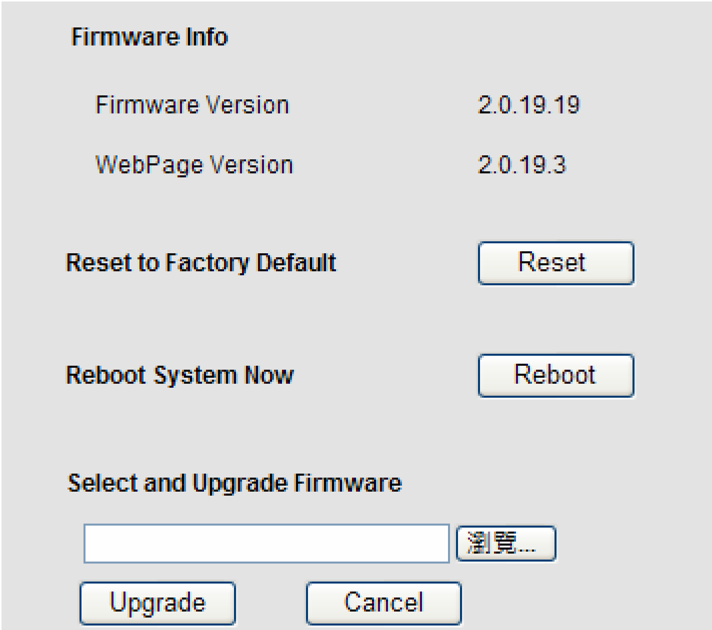


Figure 23. Firmware information page

Click the “**Browse**” button in the right side of the File Location or you can type the correct path and the filename in File Location blank.

Select the correct file you want to download to the IP Phone then click the “Upgrade” button.

Advanced settings

This page defines the Auto Provision and Remote Maintenace servers setting, it’s provide server’s IP address or domain name automatically when IP phone starts

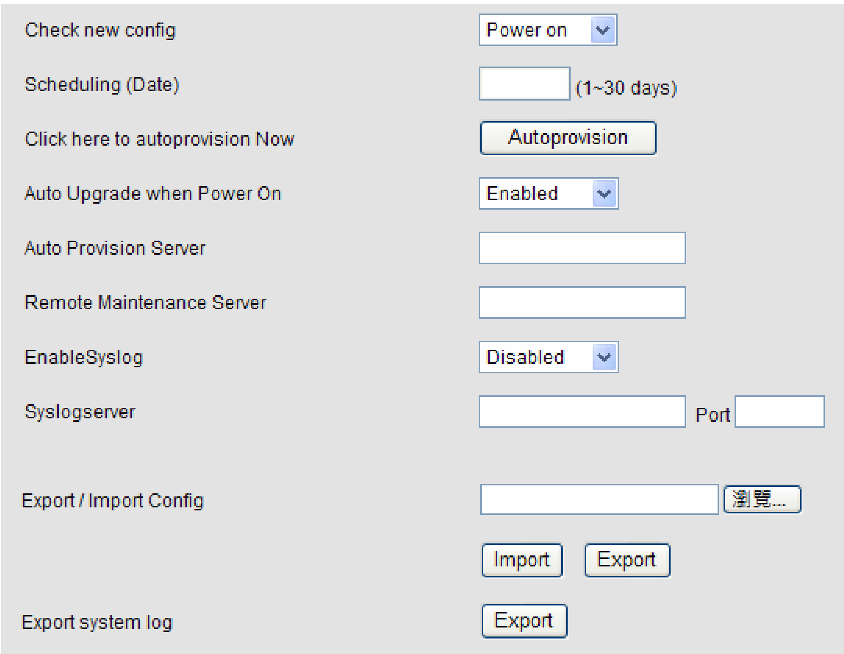


Figure 24. Auto provision setting page

Field	Description
Check New config	The device will according to the below ways to check.the new configuration. **- Power On (+ Scheduling)**: The machine will check the new firmware when power on and following the scheduling date and time. **- Scheduling**: The machine will follow the scheduling date and time to check the new firmware.
Scheduling (Data)	The machine will check the new configuration between the time range by random.
Autoprovision Now	Recheck new configuration immediately.
Auto Upgrade when Power On	When you set yes, it will auto update the firmware when power on. The default is enabled.
Auto Provision Server	Auto Provision Server's IP address or Domain name provided by ISP.
Remote Maintenance Server	Remote Maintenance Server's IP address or Domain name provided by ISP.

Security Settings

Advanced user could change the login username and the password in this page. This "**Enable Change Account**" parameter defines whether enable user to change the registered account.

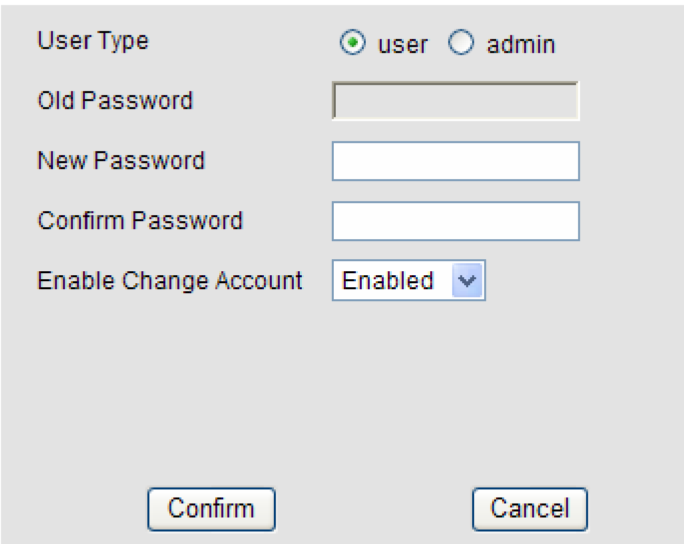


Figure 25. Security setting page

Appendix A Voice communications

There are several ways to make calls to desired destination in IP Phone. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

Case 1: Voice communication via SIP proxy server SIP-50

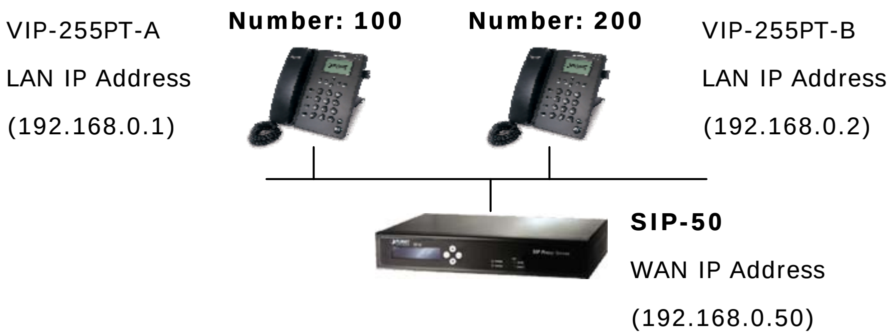


Figure 26.. Installation example with SIP-50

Machine configuration on the VIP-255PT:

STEP 1:

Log in SIP-50 and create two testing accounts/password: **100** / **123** (for VIP-255PT-A), and **200** / **123** (for VIP-255PT-B) for the voice calls.

STEP 2:

Please log in VIP-255PT-A via web browser, browse to the **Account setting** menu and. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET SIP-50 as the SIP Proxy server for SIP account, call authentications), and then the sample configuration screen is shown below:

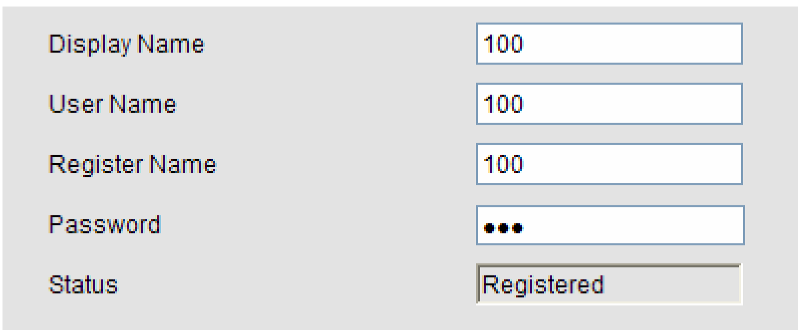


Figure 27. Web page of VIP-255PT

STEP 3:

Then browse to the **Server setting** menu and. In the setting page, please insert the SIP-50 IP address information obtained from your service provider

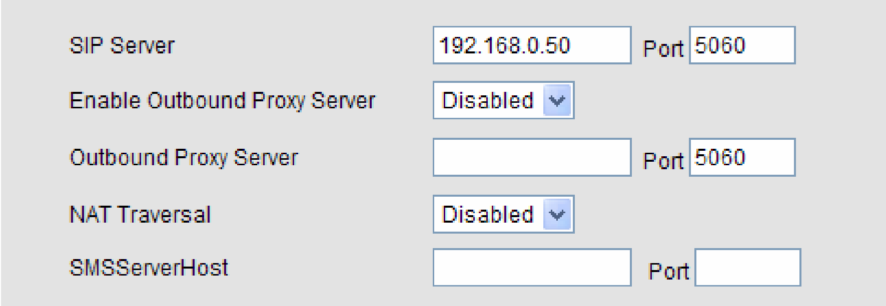


Figure 28. Web page of VIP-255PT

STEP 4:

Repeat the same configuration steps on VIP-255PT-B, and check the machine registration status, make sure the registrations are completed.

STEP 5:

To verify the VoIP communication, please pick up the telephone. Dial the destination number to make call between SIP clients. For example, VIP-255PT-A (with number 100) with keypad number 200 to VIP-255PT-B, or reversely makes calls from SIP client (VIP-255PT-B) to the number 100 (VIP-255PT-A).

Case 2: Call Forward Feature Example

In the following samples, we'll introduce the Call Forward Feature applications.

In this example, there are three VIP-255PT register to IPX-300 and VIP-255PT_A had set Call Forward function to VIP-255PT_B. (The detail registration settings of IPX-300 and VIP-255PT please refer to the instruction of Case 3)

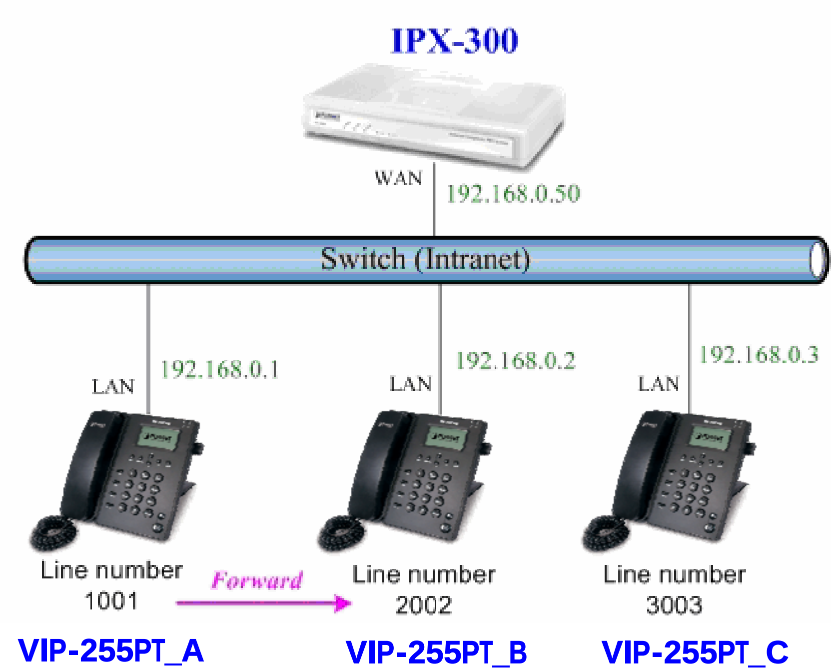


Figure 29. Installation example with IPX-300

Machine configuration on the VIP-255PT:

STEP 1:

Please log in VIP-255PT_A via web browser, browse to the **Phone Function** setting menu. In the setting page, please enable the **Always Forward** function and fill in the **Number** of VIP-255PT_B, then the sample configuration screen is shown below:

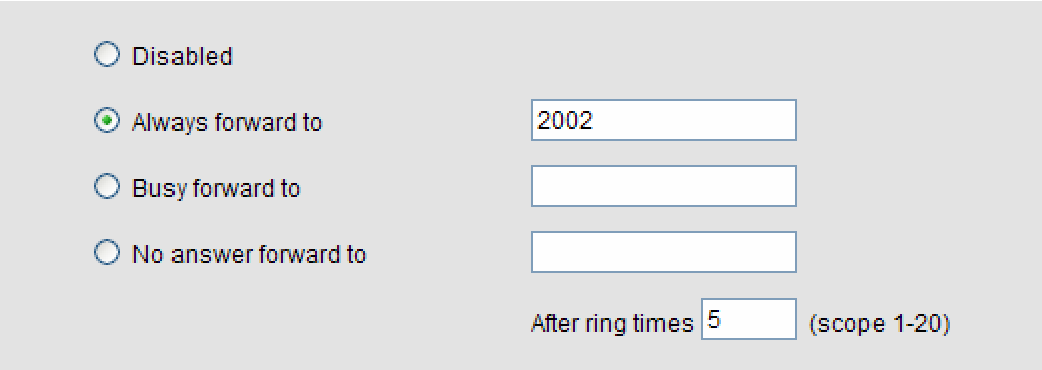


Figure 30. Web page of VIP-255PT

STEP 2:

After set up completed, it will show the always forward icon +[+ on the LCD screen.

Test the scenario:

VIP-255PT_C pick up the telephone and dial the number 1001(VIP-255PT_A), because VIP-255PT_A had set up **All Forward** function to the number 2002(VIP-255PT_B), so the number 2002(VIP-255PT_B) will ring up then it pick up the telephone and communication with the number 3003(VIP-255PT_C).

Appendix B The method of operation guide

In this section, we'll introduce the features method of operation, and lead you step by step to establish these features.

Call Transfer

A. Blind Transfer

1. B call to A and they are in the process of conversation.
2. A carry the transfer function out (Press "**FLASH**" button) to hold the conversation with B.
3. A will be hear the dial tone then input the number of C (Follow by the "#" key).
4. A will be hear the ring back tone then hung up the handset
5. C will ring up
6. C picks up the handset and conversation with B.

B. Blind Transfer

1. B call to A and they are in the process of conversation.
2. A carry the transfer function out (Press "**FLASH**" button) to hold the conversation with B,
3. A will be hear the dial tone then input the number of C (Follow by the "#" key).
4. C will ring up.
5. C picks up the handset and conversation with A.
6. A hang up and C conversation with B.

3-Way Conference

1. A and B are in the process of conversation.
2. A want to invite C to join their conversation.
3. A press "**CONF**" button to hold the conversation with B, and input the number of C (Follow by the "#" key).
4. C will be ring and entry into the 3-Way conference after C pick up the handset.

Call Waiting

1. A and B are in the process of conversation.
2. C call to A and A will hear the prompt sounds.
3. A press "**Hold**" button to hold the conversation with B, and switch to conversation with C.
4. User could also utilize the "▲" and "▼" keys to switch the communication.

Do Not Disturb

All incoming calls will be rejected.

1. Press Hold key to start this function.
2. Press Hold key or hang up to cancel this function.

Mute the Call

During a call, press **Del** key to mute your microphone. To cancel the Mute function, press the **Del** key again.

Appendix C Frequently Asked Questions List

Q : I can not register to the server？
A : 1. Check the IP address. If you set your LAN port in DHCP mode, please make sure that your DHCP server is on. 2. Check your gateway. 3. Check your DNS server. 4. Make sure your account information is the same as you have got from your ISP. 5. Check whether the SIP server is on. 6. Check the SIP register port, the default value is 5060.
Q : I can't get the IP address?
A : 1.Make sure you have plugged the Ethernet cable into the LAN port. 2. Make sure that the DHCP server is on, and there are available IP addresses in the server. 3. Try to set your WAN port to static IP client mode.
Q : During a call, I can not hear any voice?
A : 1.Make sure Your handset is tightly connected with the phone. 2. Check whether you have muted the conversation or not. 3. Consult the outbound server details with your ISP.
Q : Have DTMF problem?
A : 1.Check which kind of DTMF you are using, and whether it is compatible with the server 2.Consult the payload value with your ISP
Q : How to change the time?
A : Select the time zone on the webpage. **Note:** You can't change the time manually because that our phone will automatically get the time from the SNTP server.
Q : **How to answer the incoming calls during a call?**
A : If a call comes in when you are in a conversation, press the HOLD button to answer the incoming call.
Q : **How to refuse incoming calls during a call?**
A : You can turn off the function of call waiting, and then our phone will refuse all the incoming calls when you are in a conversation.
Q : **How to send SMS?**
A : You could edit the SMS in the MENU-> Messages->Text Messages. **Note:** Make sure that the SIP server you have registered supports SMS function.

Q：**How to update the firmware?**
A : 1. Update the firmware on the webpage Upgrade-> Select and Upgrade Firmware. 2. Select the correct file you want to download to the IP Phone then click the “Upgrade” button.
Q：**How to auto provision?**
A : Consult the auto provision server address with your ISP.
Q：**How to adjust volume?**
A : During a call, press ▼/▲ key to adjust the volume of earpiece or speaker.
Q：**How to select ring?**
A : 1.There are four kinds of ring styles to choose. 2. To adjust the ring volume, please press ▼/▲ key on the phone.

Appendix D VIP-255PT Specifications

Product	SIP PoE IP Phone
Model	VIP-255PT
Hardware	
LAN	1 x 10/100Mbps RJ-45 port Power Over Ethernet 802.3af compliant
PC	1 x 10/100Mbps RJ-45 port
LCD display	132 x 64 dot matrix graphic LCD
Speaker	Full duplex hands free speaker phone
Protocols and Standard	
Standard	SIP 2.0 (RFC3261), MD5 for SIP authentication (RFC2069/ RFC 2617), SIP outbound proxy, SIP NAT Traversal Support STUN (RFC3489)
Voice codec	G.711: 64k bit/s (PCM) G.723.1: 6.3k / 5.3k bit/s G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) G.729A: 8k bit/s (CS-ACELP) G.729B: adds VAD & CNG to G.729
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Acoustic echo canceller (AEC) G.165: Line echo canceller (LEC) Jitter Buffer
Supplementary services	Caller ID 3-way conference Immediate (unconditional) call forwarding Busy call forwarding No answer calls forwarding Call Hold/Waiting/Transferring
Call history	Record incoming call Outgoing call Missed (not accepted) call history
Protocols	SIP v1 (RFC2543), v2(RFC3261), TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, RARP, DNS, DHCP, SNTP, PPPoE
Network and Configuration	
Access Mode	Static IP, PPPoE, DHCP
Management	Web, LCD menu keypad, auto-provision by TFTP/FTP/HTTP
Dimension (W x D x H)	184 mm x 200 mm x 48 mm
Operating Environment	0~50 degree C, 0~90% humidity
Power Requirement	5V DC, 1A
EMC/EMI	CE, FCC Class B